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| APPLICATION NO.  | FILING DATE | FIRST NAMED INVENTOR | ATTORNEY DOCKET NO.          | CONFIRMATION NO.       |
|--|-------------|----------------------|------------------------------|------------------------|
| 10/529,280   | 08/24/2005  | Yaakov Stein         | RAD 19.779<br>(100757-00082) | 4342                   |
| 26304 7590 02/12/2009<br>KATTEN MUCHIN ROSENMAN LLP<br>575 MADISON AVENUE<br>NEW YORK, NY 10022-2585 |             |                      | EXAMINER<br>HE, JIALONG      |                        |
|  |             |                      | ART UNIT<br>2626             | PAPER NUMBER           |
|  |             |                      | MAIL DATE<br>02/12/2009      | DELIVERY MODE<br>PAPER |

**Please find below and/or attached an Office communication concerning this application or proceeding.**

The time period for reply, if any, is set in the attached communication.

|                              |                                      |                                      |  |
|------------------------------|--------------------------------------|--------------------------------------|--|
| <b>Office Action Summary</b> | <b>Application No.</b><br>10/529,280 | <b>Applicant(s)</b><br>STEIN, YAAKOV |  |
|                              | <b>Examiner</b><br>JIALONG HE        | <b>Art Unit</b><br>2626              |  |

**-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --**

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

- 1) ☒ Responsive to communication(s) filed on 29 December 2008.
- 2a) ☒ This action is **FINAL**.                      2b) ☐ This action is non-final.
- 3) ☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

- 4) ☐ Claim(s) 1-18 is/are pending in the application.
- 4a) Of the above claim(s) \_\_\_\_\_ is/are withdrawn from consideration.
- 5) ☐ Claim(s) \_\_\_\_\_ is/are allowed.
- 6) ☒ Claim(s) 1-18 is/are rejected.
- 7) ☐ Claim(s) \_\_\_\_\_ is/are objected to.
- 8) ☐ Claim(s) \_\_\_\_\_ are subject to restriction and/or election requirement.

**Application Papers**

- 9) ☐ The specification is objected to by the Examiner.
- 10) ☐ The drawing(s) filed on \_\_\_\_\_ is/are: a) ☐ accepted or b) ☐ objected to by the Examiner.  
Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).  
Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).
- 11) ☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

**Priority under 35 U.S.C. § 119**

- 12) ☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
- a) ☐ All    b) ☐ Some \*    c) ☐ None of:
1. ☐ Certified copies of the priority documents have been received.
2. ☐ Certified copies of the priority documents have been received in Application No. \_\_\_\_\_.
3. ☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

\* See the attached detailed Office action for a list of the certified copies not received.

**Attachment(s)**

- |  |   |
|--|---|
| 1) <input checked="" type="checkbox"/> Notice of References Cited (PTO-892)          | 4) <input type="checkbox"/> Interview Summary (PTO-413)           |
| 2) <input type="checkbox"/> Notice of Draftsperson's Patent Drawing Review (PTO-948) | Paper No(s)/Mail Date. _____                                      |
| 3) <input type="checkbox"/> Information Disclosure Statement(s) (PTO/SB/08)          | 5) <input type="checkbox"/> Notice of Informal Patent Application |
| Paper No(s)/Mail Date _____  | 6) <input type="checkbox"/> Other: _____                          |

## **DETAILED ACTION**

### ***Examiner Change***

1. It should be note the Examiner was changed due to the previous examiner being no longer available.

### ***Response to Amendment***

2. Applicant's amendment filed on 12/29/2008 has been entered. Claims 1, 3, 12, 17 have been amended. Claims 19-23 have been cancelled. No claim has been added. Claims 1-18 are still pending in this application, with claims 1, 12 and 17 being independent.

The specification was objected in previous office action due to failing to provide proper antecedent basis for the claimed subject matter. The applicant cancelled claims 19-23. The objection to specification is withdrawn.

### ***Response to Arguments***

3. Applicant's arguments have been fully considered but they are not persuasive for the following reasons.

Regarding rejection to independent claims 1, 12 and 17 under 102(e), the applicant amended these claims by adding new limitations "separately encapsulating

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speech and music components along with synchronization information into respective packets suitable for transmission over a packet network” and argued that Saunders fails to teach this new limitation.

The examiner note that Saunders discloses both speech and non-speech audio (music) data streams are synchronized with video frames (**col. 16, lines 20-26, frame is a packet**) and CODEC can be AC-3 (**col. 20, lines 1-5**) or MPEG-4 AAC (**col. 15, lines 8-15, also fig. 8**). It is known AC-3 standard contains multiple audio channels (audio/voice or different languages) and the compressed data are packetized with synchronization information. The AC-3 standard defines a transport protocol in which the packets are suitable for transmission over a packet network (**See AC-3 standard (a reference cited but not relied upon); page 12-13, bit stream syntax, it shows each packet (syncframe) has synchronization header followed by 6 audio blocks; page 107, stream is packetized into PES packets**). Therefore, Saunders **implicitly discloses** the limitation: encapsulating speech and music components along with synchronization into respective packets suitable for transmission over a packet network.

### ***Claim Objections***

4. Claims 1, 9, 12, 14 and 17 are objected to because of the following informalities:

Claims 1, 12 and 17 recites a new limitation "speech packet suitable for transmission over a packet network". The term "suitable for" is not a positive limitation (see MPEP 2111.04) . The Examiner interprets prior art to read on this limitation when speech packet can be transferred over a packet network (such as the Internet).

Claims 9 and 14 recite "MDI-encoder" which appears to be a misspelling of the word "MIDI-encoder".

### ***Claim Rejections - 35 USC § 112***

5. The following is a quotation of the second paragraph of 35 U.S.C. 112:

The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

6. Claim 8, 15 and 18 are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

Claims 8, 15 and 18, which depends on claim 1, 12 and 17, respectively, recite the limitation "said embedded synchronization". Since "embedded synchronization" has been deleted from their corresponding independent claims. There is insufficient antecedent basis for this limitation in the claim.

Claims 12 and 18 have the similar problem as claim 8.

***Claim Rejections - 35 USC § 102***

7. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

A person shall be entitled to a patent unless –

(e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

8. Claims 1, 5, 6, 10-13, and 16-17 are rejected under 35 U.S.C. 102 (e) as being clearly anticipated by Saunders (US Pat. 6,351,733, hereinafter to as Saunders).

Regarding claim 1, Saunders discloses a system providing low bit-rate compression of data comprising speech and music components for transmission, over a network, said system comprising:

a. a speech encoder encoding said speech component via a first encoding algorithm (**Fig. 9-12, col. 15, lines 21-24, signal is compressed with a speech-only codec**) and encapsulating said encoded speech component along with synchronization information (**col. 13, lines 55-65, using compression algorithms designed specially for speech signals, time synchronized with the other audio tracks that are compressed using more general audio compression algorithm**).

b. a music encoder encoding said music component via a second encoding algorithm, said second encoding algorithm different from said first encoding algorithm (**Fig. 9-12, col. 15, lines 21-30; non-speech signal is encoded using a general compression algorithm; col. 16, lines 9-12, two parallel streams are fed into two distinct compression algorithms**) and encapsulating said encoded music component along with synchronization information (**col. 16, lines 15-25, a general audio compression unit time-synchronized with the voice-only compression**)

wherein said first and second encoding algorithms are chosen to allow for low bit-rate compression of speech and music respectively (**col. 15, lines 27-30, the distinction between compression using the speech-only codec and the general codec helps to reduce the required bandwidth**).

Saunders discloses both speech and non-speech audio (music) data streams are synchronized with video frames (**col. 16, lines 20-26, frame is a packet**) and CODEC can be AC-3 (**col. 20, lines 1-5**) or MPEG-4 AAC (**col. 15, lines 8-15, also fig. 8**). It is known AC-3 standard contain multiple data streams (audio or voice) and the compressed data are packetized with synchronization information. The AC-3 standard a transport protocol in which the packets are suitable for transmission over a packet network (**See AC-3 standard, cited but not relied upon**). Therefore, Saunders **implicitly discloses** encapsulating speech and music components along with synchronization into respective packets suitable for transmission over a packet network.

With respect to claim 5, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches a system as per claim 1, wherein audio volumes associated with said speech component and said music component are modifiable relative to each other (**col. 6, lines 44-54, the volume of each signal may be independently adjusted; col. 22, lines 53-67; claim 9; the ratio between the levels of the channels may be adjusted**).

With respect to claim 6, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches a system as per claim 1, wherein said speech encoder is a LPC, MELP, CELP, or waveform interpolation encoder (**col. 15, lines 24-27, speech coding can be conducted using any known speech codec such as the CELP codec**).

With respect to claim 10, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches a system as per claim 1, wherein said music encoder is a transform-based encoder (**col. 20, lines 20-51, audio codecs which may be used include MP3, which is a transform-based encoder**).

With respect to claim 11, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches a system as per claim 1, wherein said network is any of the following: local area network, wide area network, the Internet,



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cellular network, storage network, or wireless network (**col. 9, lines 37-40; col. 25, lines 23-30; transmission may be an ISDN transmission to a PC modem**).

With respect to independent claim 12, Saunders teaches a system providing low bit-rate compression of audio comprising speech and music components (**col. 2, lines 15-18, audio program is produced so that the audio content is readily fabricated for transmission**), said system comprising:

a. a first analog-to-digital converter converting said speech component into a digital speech signal (**col. 8, lines 16-43; all audio sources utilize microphones to transducer audio information into real-time electrical signals, from which digital masters are created, using a digital format such as PCM**);

b. a speech encoder encoding said digital speech signal via a first encoding algorithm (**Fig. 9-12, col. 15, lines 21-24, signal is compressed with a speech-only codec**);

d. a second analog-to-digital converter converting said music component into a digital music signal (**col. 8, lines 16-43; all audio sources utilize microphones to transducer audio information into real-time electrical signals, from which digital masters are created, using a digital format such as PCM**);

e. a music encoder encoding said digital music signal via a second encoding algorithm, said second encoding algorithm different from said first encoding algorithm (**Fig. 9-12, col. 15, lines 21-30; non-speech signal is encoded using a general**

**compression algorithm; col. 16, lines 9-12, two parallel streams are fed into two distinct compression algorithms); and**

g. a multiplexer multiplexing said outputs of said speech audio formatter and said music audio formatter for transmission over said channel (**Fig. 12, col. 16, lines 24-28, outputs of compression units are multiplexed in a specific manner so that the audio can be transmitted**).

Saunders discloses both speech and non-speech audio (music) data streams are synchronized with video frames (**col. 16, lines 20-26, frame is a packet**) and CODEC can be AC-3 (**col. 20, lines 1-5**) or MPEG-4 AAC (**col. 15, lines 8-15, also fig. 8**). It is known AC-3 standard contain multiple data streams (audio or voice) and the compressed data are packetized with synchronization information. The AC-3 standard a transport protocol in which the packets are suitable for transmission over a packet network (**See AC-3 standard, cited but not relied upon**). Therefore, Saunders **implicitly discloses** encapsulating speech and music components along with synchronization into respective packets suitable for transmission over a packet network.

With respect to claim 13, Saunders teaches everything claimed, as applied above (see claim 12); in addition, Saunders further teaches a system as per claim 12, wherein said speech encoder is a LPC, MELP, CELP or waveform interpolation encoder (**col. 15, lines 24-27, speech coding can be conducted using any known speech codec such as the CELP codec**).

With respect to claim 16, Saunders teaches everything claimed, as applied above (see claim 12); in addition, Saunders further teaches a system as per claim 12, wherein said music encoder is a transform-based encoder (**col. 20, lines 20-51, audio codecs which may be used include MP3, which is a transform-based encoder**).

Claim 17 is a method claim and is similar to claim 12. Therefore, claim 17 is rejected based on the same rationale.

### ***Claim Rejections - 35 USC § 103***

9. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negated by the manner in which the invention was made.

10. Claim 2 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Kane et al. (US 5,293,450 hereinafter referred to as Kane).

With respect to claim 2, Saunders teaches everything claimed, as applied above (see claim 1); but Saunders does not explicitly teach a system as per claim 1, wherein said data is a composite of said speech and music components and said system further comprises a signal separator, said signal separator separating said speech and music

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components from said composite. However, the examiner contends that this concept was well known, as taught by Kane.

In the same field of vocal audio coding, Kane teaches a system to code the voice signal separately from the noise (non-voice) signal (Fig. 7, circuit 31-33 code the noise). Thus a voice may be separated from background music and both coded separately (col. 5, line 64-col. 6, line 3).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the system of Saunders with the voice/non-voice separator of Kane, in order to increase the functionality of the system to be able to use audio data that does not already have vocal data separated from other audio data.

11. Claim 3 and 4 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of in view of Tsukagoshi (US 6,104,861, hereinafter referred to as Tsukagoshi) and further in view of Ito et al. (US 2001/0012444, hereinafter referred to as Ito).

With respect to **claim 3**, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches that the audio may be synchronized with associated video (*col. 10, lines 29-31; col. 16, lines 21-24*) such as a motion picture, but Saunders does not explicitly teach a system as per claim 1, wherein said data further comprises a text component, a video component, and a graphics component, said system further comprising:

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- a text formatter transforming said text component into a format suitable for transmission and embedding synchronization information associated with said text component;
- a video encoder encoding said video component via a third encoding algorithm, said third encoding algorithm different from said first and second encoding algorithms; transforming said encoded video signal into a format suitable for transmission; and embedding synchronization information associated with said video component;
- a graphics encoder encoding said graphics component via a fourth encoding algorithm, said fourth encoding algorithm different from said first, second, and third encoding algorithms; transforming said encoded graphics into a format suitable for transmission; and embedding synchronization information associated with said graphics component; and
- said multiplexer in (c) additionally multiplexing the output of said text formatter, said video encoder, and graphics encoder.

However, the examiner contends that these concepts were well known in the art, as taught by Tsukagoshi and Ito.

In the same field of endeavor of encoding and decoding of multiple types of data streams, Tsukagoshi teaches a system for encoding a video broadcast into multiple types of data streams for transmission. The system includes: a subtitle encoder (*Fig. 9A, element 57*) that generates

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subtitles (*col. 13, lines 16-29*), selects data for compression and encodes it (*col. 13, lines 30-55*), and forwards the subtitle information to the multiplexer for multiplexing with the audio and video data to be transmitted (*col. 14, lines 33-44*); a video encoder (*Fig. 9A, element 52*) which compresses digital video for video transmission (*col. 13, lines 1-8*) and synchronized with the subtitle data (*col. 13, lines 5-8*) and multiplexed with the other data by the multiplexer (*col. 14, lines 33-44*); as well as an audio encoder (*Fig. 9A, lines 9-15*).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to use the audio encoding method of Saunders in the data encoding method of Tsukagoshi because Tsukagoshi teaches an efficient way of transmitting video broadcast comprised of multiple data streams, as suggested by Saunders (*col. 10, lines 29-31; col. 16, lines 21-24*).

In the same field of endeavor of encoding and decoding of multiple types of data streams, Ito teaches a system for encoding a broadcast for transmission. The system comprises separate speech encoder (*Fig. 18, element 5001*) and a character object encoder (*Fig. 18, element 5004*), as well as a synthesized image object encoder (*Fig. 18, element 5003, paragraphs [0087-0088]*). The output of the encoder is multiplexed with the outputs of the other encoders and output as a bitstream for transmission (*Fig. 18*). A synthesized image may be a computer graphic (*paragraph [0088]*) such as a background object or weather information image (*paragraph [0257]*).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify the system of Saunders and Tsukagoshi with the additional graphic encoder of Ito, because the addition of graphics to the video would enable more information to

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be represented graphically to viewers, especially in such applications as a news program  
(*paragraph [0257]*).

With respect to **claim 4**, Saunders in view of Tsukagoshi and Ito teach everything claimed, as applied above (see claim 3); in addition, Saunders does not but Tsukagoshi does teach a system as per claim 3, wherein said text component corresponds to subtitles associated with said video components (*Fig. 9A, col. 13, lines 16-29, information from character generator that is encoded is subtitles*).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to use the audio encoding method of Saunders in the data encoding method of Tsukagoshi because Tsukagoshi teaches an efficient way of transmitting video broadcast comprised of multiple data streams, as suggested by Saunders (*Saunders, col. 10, lines 29-31; col. 16, lines 21-24*).

12. Claim 7 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of in view of Newlin (US 5,774,857).

With respect to claim 7, Saunders teaches everything claimed, as applied above (see claim 1); but Saunders does not teach a system as per claim 1, wherein said speech encoder is used in conjunction with a speech-to-text converter, and

- said speech-to-text converter converting said speech component to a text component; and

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- said speech encoder encoding said text components and formatting said encoded text into a format suitable for transmission.

However, the examiner contends that these concepts were well known in the art, as taught by Newlin.

In the same field of endeavor of encoded audio and video transmission, Newlin teaches a system that accepts the input of an audio signal (Fig. 3, col. 11, lines 45-48) and processes the audio signal with a speech recognition subsystem to form a text representation of speech (Fig. 3, element 307, col. 11, lines 51-54). The text is then processed by the closed caption encoder (Fig. 3, element 311) to convert it to a closed caption video format (col. 11, lines 54-61) to prepare it for mixing with the video signal for transmission (col. 11, lines 61-67).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the system of Saunders with the speech-to-text converter of Newlin, in order to automatically generate a visual representation of speech without use of dedicated systems or user entry of the material to provide access for the hearing impaired (Newlin, col. 2, line 4-15).

13. Claim 8, 15, and 18 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Holmes et al. (US 5,506,932, hereinafter referred to as Holmes).



With respect to **claim 8**, Saunders teaches everything claimed, as applied above (see claim 1); in addition, Saunders teaches that audio tracks are time-synchronized and video-frame synchronized (*col. 16, lines 15-24; claim 19*) but Saunders does not specifically teach a system as per claim 1, wherein said embedded synchronization information is any of the following: timestamps, synchronization labels, media synchronization tags, synchronizing tokens, or wait-on-event commands. However, the examiner contends that this concept was well known in the art, as taught by Holmes.

In a related field of endeavor of audio and video synchronization, Holmes teaches that audio data is synchronized to the video data on a frame-by-frame basis using a clock (*col. 5, lines 20-23*). Each frame is identified by a time stamp (*col. 6, lines 23-24*).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to implement the synchronization taught by Saunders with the method using a time stamp as taught by Holmes, because the time stamp method is one of a finite number of methods known to be useful for audio-video data stream synchronization, as a person with ordinary skill has good reason to pursue the known options within his or her technical grasp. Using the known method would have had predictable results, so it would have been obvious to try the time stamp synchronization method to synchronize the data.

With respect to **claim 15**, Saunders teaches everything claimed, as applied above (see claim 12); in addition, Saunders teaches that audio tracks are time-synchronized and video-frame synchronized (*col. 16, lines 15-24; claim 19*) but Saunders does not specifically teach a system as per claim 12, wherein said embedded synchronization

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information is any of the following: timestamps, synchronization labels, media synchronization tags, synchronizing tokens, or wait-on-event commands. However, the examiner contends that this concept was well known in the art, as taught by Holmes.

In a related field of endeavor of audio and video synchronization, Holmes teaches that audio data is synchronized to the video data on a frame-by-frame basis using a clock (*col. 5, lines 20-23*). Each frame is identified by a time stamp (*col. 6, lines 23-24*).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to implement the synchronization taught by Saunders with the method using a time stamp as taught by Holmes, because the time stamp method is one of a finite number of methods known to be useful for audio-video data stream synchronization, as a person with ordinary skill has good reason to pursue the known options within his or her technical grasp. Using the known method would have had predictable results, so it would have been obvious to try the time stamp synchronization method to synchronize the data.

With respect to **claim 18**, Saunders teaches everything claimed, as applied above (see claim 17); in addition, Saunders teaches that audio tracks are time-synchronized and video-frame synchronized (*col. 16, lines 15-24; claim 19*) but Saunders does not specifically teach a method as per claim 17, wherein said embedded synchronization information is any of the following: timestamps, synchronization labels, media synchronization tags, synchronizing tokens, or wait-on-event commands. However, the examiner contends that this concept was well known in the art, as taught by Holmes.

In a related field of endeavor of audio and video synchronization, Holmes teaches that audio data is synchronized to the video data on a frame-by-frame basis using a clock (*col. 5, lines 20-23*). Each frame is identified by a time stamp (*col. 6, lines 23-24*).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to implement the synchronization taught by Saunders with the method using a time stamp as taught by Holmes, because the time stamp method is one of a finite number of methods known to be useful for audio-video data stream synchronization, as a person with ordinary skill has good reason to pursue the known options within his or her technical grasp. Using the known method would have had predictable results, so it would have been obvious to try the time stamp synchronization method to synchronize the data.

14. Claim 9 and 14 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Rabowsky et al. (US 5,680,512, hereinafter referred to as Rabowsky).

With respect to **claim 9**, Saunders teaches everything claimed, as applied above (see claim 1); but Saunders does not specifically teach a system as per claim 1, wherein said music encoder is a MIDI-encoder or linear musical score notation. However, the examiner contends that this concept was well known in the art, as taught by Rabowsky.

In the same field of endeavor of audio coding, Rabowsky teaches separate encoding of a voice and non-voice signal, which are combined to form the audio signal (*col.1, lines 49-60*).

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The encoder for the non-voice signal (*Fig. 1, element 12*) may use MIDI (*col. 5, lines 19-25, signal is applied to the encoder; col. 5, lines 43-57, MIDI elements are stored for reference in the encoder*).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the audio coding system of Saunders with the MIDI encoder of Rabowsky, in order to increase the functionality of the coder to encode MIDI data.

With respect to **claim 14**, Saunders teaches everything claimed, as applied above (see claim 12); but Saunders does not specifically teach a system as per claim 12, wherein said music encoder is a MDI-encoder or linear musical score notation. However, the examiner contends that this concept was well known in the art, as taught by Rabowsky.

In the same field of endeavor of audio coding, Rabowsky teaches separate encoding of a voice and non-voice signal, which are combined to form the audio signal (*col.1, lines 49-60*). The encoder for the non-voice signal (*Fig. 1, element 12*) may use MIDI (*col. 5, lines 19-25, signal is applied to the encoder; col. 5, lines 43-57, MIDI elements are stored for reference in the encoder*).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the audio coding system of Saunders with the MIDI encoder of Rabowsky, in order to increase the functionality of the coder to encode MIDI data.

***Conclusion***

15. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

\* ATSC Standard: Digital Audio Compression (AC-3) --- discloses compressed audio data are packetized with synchronization information in each packet.

\* Tanaka; Yoshiaki et al. (US 6757659 B1) - audio signal processing apparatus.

\* Michener; James A. (US 7283965 B1) - delivery and transmission of dolby digital ac-3 over television broadcast.

16. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL**. See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire **THREE MONTHS** from the mailing date of this action. In the event a first reply is filed within **TWO MONTHS** of the mailing date of this final action and the advisory action is not mailed until after the end of the **THREE-MONTH** shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of

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the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

17. Any inquiry concerning this communication or earlier communications from the examiner should be directed to JIALONG HE whose telephone number is (571)270-5359. The examiner can normally be reached on Monday-Thursday, 7:00 - 4:30, Alt Friday, EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Patrick Edouard can be reached on (571) 272-7603. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see <http://pair-direct.uspto.gov>. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/JH/

/Patrick N. Edouard/  
Supervisory Patent Examiner, Art Unit 2626